Benchmark ADC1
Instruction Manual

2-Channel 24-bit 192-kHz
Audio Analog-to-Digital Converter

Federal Communications Commission (FCC) Notice (U.S. Only)

NOTICE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the FCC rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference.
2. This device must accept any interference received including interference that may cause undesired operation.

Instructions to Users: This equipment complies with the requirements of FCC (Federal Communication Commission) equipment provided that following conditions are met:

1. XLR Digital Output: Shielded 110-Ohm AES/EBU digital audio cable with connector shell bonded to shield must be used.
2. BNC Digital Connections: Shielded 75-Ohm coaxial cable must be used.

NOTICE: Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Safety Information

Do NOT service or repair this product unless properly qualified. Only a qualified technician should perform repairs.

For continued fire hazard protection, fuses should be replaced ONLY with the exact value and type as indicated on the rear panel.

Do NOT substitute parts or make any modifications without the written approval of Benchmark Media Systems, Inc. Doing so may create safety hazards and void the warranty.

Contents

Overview

The ADC1 is a reference-quality, 2-channel 192-kHz 24-bit audio analog-to-digital converter featuring Benchmark's *UltraLock*™ technology. The ADC1 is designed for maximum transparency. It is well suited for the most demanding applications in studios and mastering facilities. A rugged and compact half-wide 1 RU enclosure also makes the ADC1 an excellent choice for location recording, broadcast facilities, and mobile rigs. The internal power supply supports all international voltages and has generous margins for over and under voltage conditions.

The ADC1 achieves outstanding performance over a wide range of input levels. Each channel has a 41-detent variable gain control, a 10-turn calibration trimmer, and a 3-position first-stage gain switch (0, 10, and 20 dB). Each channel has a two-position toggle switch that selects either the 41-detent pot or the 10-turn trimmer. Both the pot and the trimmer have a 20 dB adjustment range. In combination with the first-stage gain switch, these controls provide exceptional SNR and THD+N performance over a 40 dB adjustment range. The 10-turn calibration trimmer may be used to calibrate the ADC1 to precise studio reference levels. It may also be used to optimize the gain-staging between a microphone preamplifier and the ADC1.

The ADC1 has four digital outputs (1 balanced XLR, 2 coaxial, and 1 optical). The optical output supports AES, ADAT, and ADAT S/MUX. The two coaxial outputs (Main and Aux) can operate simultaneously at different word lengths and even at different sample rates. The ADC1 has the flexibility to allow simultaneous high-resolution and low-resolution recordings. For example, the main outputs of the ADC1 can be set to 192 kHz 24-bits while the auxiliary output is set to 44.1 kHz 16-bits for a safety backup or CDR demo recording. Both the Main and Aux Outputs originate from the same A/D converter. All outputs are professional format.

The ADC1 has a Word Clock output that follows the sample rate of the Main Outputs. The Word Clock output is active in all modes of operation.

A multi-format clock input automatically recognizes AES/EBU, SPDIF, Word Clock, or Super Clock signals. This clock input is used to synchronize the Main Outputs. If desired, the Main Outputs may be driven from internal sources. The ADC1 will automatically revert to an internal clock source when the external clock is lost.

The ADC1 has two clock modes: Auto and Internal. Both modes support 44.1, 48, 88.2, 96, 176.4 and 192 kHz.

The Auto mode allows the ADC1 to lock to an external clock reference. In Auto mode, the ADC1 will follow changes in sample rate, and/or changes in the type of reference signal (AES, SPDIF, word clock, or super clock).

When a clock reference is not available, the Internal mode must be used, and a sample-rate must be selected (44.1, 48, 88.2, 96, 176.4, or 192 kHz). When the Internal mode is active, the ADC1 is acting as clock master, will only operate at the selected sample rate, and will ignore any signal at the clock reference input. If Internal mode is used, all devices connected to the ADC1 digital outputs will need to be configured to lock to the ADC1. Use the clock output on the back of the ADC1 if the connected devices require word clock.

The Benchmark UltraLock system is 100% jitter immune. The A/D conversion clock is totally isolated from the AES/EBU, SPDIF, ADAT, WC, and super clock interfaces. This topology outperforms two-stage PLL designs. In fact, no jitter-induced artifacts can be detected using an Audio Precision System 2 Cascade test set. Measurement limits include detection of artifacts as low as -140 dBFS, application of jitter amplitudes as high as 12.75 unit intervals (UI) and application of jitter over a frequency range of 2 Hz to 200 kHz. A poor-quality clock reference will not degrade the jitter performance of the ADC1. In addition, the AES/EBU receiver IC has been selected for its ability to decode signals in the presence of very high levels of jitter. The Benchmark UltraLock system delivers consistent performance under all operating conditions.

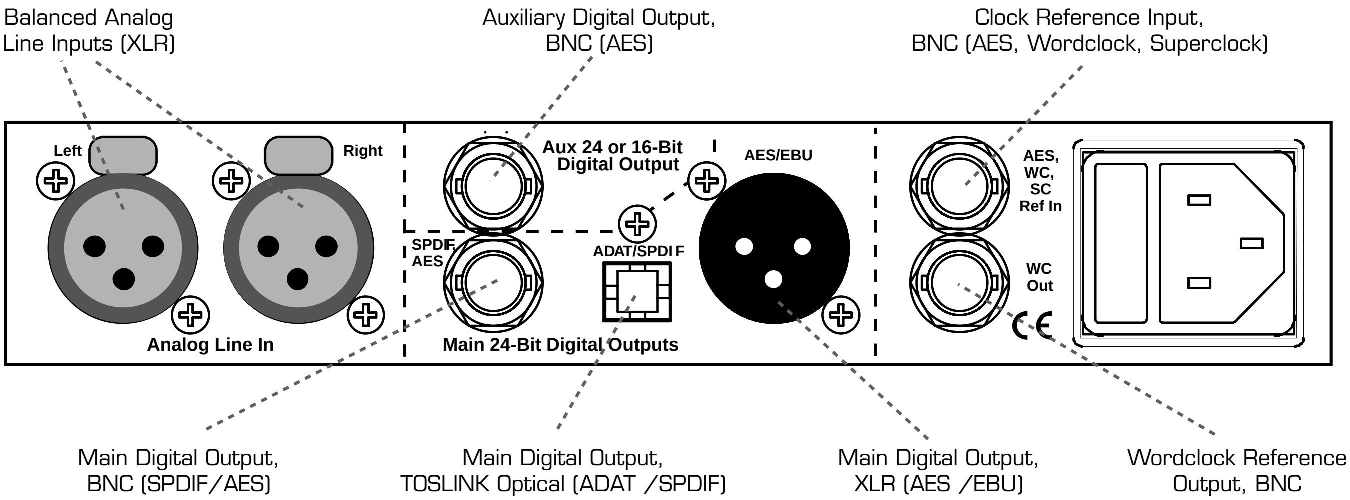
The ADC1 is designed to perform gracefully in the presence of errors and interruptions at the clock reference input. The ADC1 follows an audio-always design philosophy. Audio is present at the outputs shortly after applying power to the unit. The ADC1 will even lock to and AES/EBU signal that has its sample-rate status bits set incorrectly. Sample rate is determined by measuring the incoming signal. Lack of sample rate status bits or incorrectly set status bits will not cause loss of audio.

The ADC1 is phase accurate between channels, and between other ADC1 boxes when locked to AES/EBU or word clock reference signals. The word clock output from one ADC1 may be connected to the clock input on another ADC1 to expand the number of phase-accurate conversion channels.

Features

- Two analog-to-digital conversion channels
- Two XLR balanced analog inputs providing high-performance over a 43 dB range
- -14 dBu to +29 dBu input sensitivity range (at 0 dBFS)
- Two 0 dB, 10dB, and 20 dB first-stage gain switches (1 per channel)
- Two 41-detent gain controls with a 20 dB range (1 per channel)
- Two 10-turn gain calibration controls with a 20 dB range (1 per channel)
- Benchmark 9-segment dual-range digital LED meters
- Sample Rate LED indicators
- Conversion at 44.1, 48, 88.2, 96, 176.4, and 192 kHz
- Versatile Auto and Internal clock modes
- Multifunction clock input with auto-recognition of AES, SPDIF, Word Clock, or Super Clock
- Word Clock output
- Total jitter immunity with Benchmark's, phase-accurate *UltraLock*™ technology
- Simultaneous output at two different sample rates
- Simultaneous 16 and 24-bit outputs
- Four digital outputs (1 XLR, 2 Coax, 1 optical)
- AES/EBU, ADAT, and ADAT S/MUX2, and ADAT S/MUX4 output formats
- THD+N = -104 dB, 0.00063% @ -3 dBFS input, SNR 121 dB A-weighted
- Reliable and consistent performance under all operating conditions
- Internal 115 V, 230 V, 50-60 Hz international power supply with very wide operating range
- Low radiation toroidal power transformer significantly reduces hum and line related interference
- Meets FCC Class B and CE emissions requirements

Connections



Balanced Analog Line Inputs

Left and Right balanced inputs use locking Neutrik™ gold-pin female XLR jacks. These inputs have a wide operating range. The input sensitivity (at 0 dBFS) ranges from -20 dBu (at maximum gain) to +29 dBu (at minimum gain). The input impedance is 200k Ohms balanced, and 100k Ohms unbalanced. The high input impedance and input sensitivity, allow direct connections from many instrument pickups (adapter cable required). Direct connection of piezo pickups is not recommended as these pickups require higher input impedances (to prevent low-frequency roll-off problems).

- XLR pin 2 = + Audio In
- XLR pin 3 = - Audio In
- XLR pin 1 = Cable Shield (grounded directly to the chassis to prevent internal ground loops)

To adapt to unbalanced sources

1. Connect "+" or hot (tip on ¼ phone plug, center pin on RCA plug) to XLR pin 2.
2. Connect ground (sleeve on ¼" phone plug, case on RCA plug) to XLR pins 3 and 1.

Note it is best to used balanced wiring ("+", "-", "shield") and to tie the "-"and "shield" at the unbalanced connector.

Clock Reference Input

This input auto-detects AES/EBU, SPDIF, Word Clock, or Super Clock signals, and automatically follows changes in sample-rate. When Auto mode is active the ADC1 will lock to the external clock source. Benchmark's UltraLock circuitry isolates the conversion clock from any jitter present on the clock reference. Auto Mode will not degrade the conversion quality of the ADC1 even when very high levels of jitter are present on the clock reference.

Digital Outputs

The ADC1 has four digital audio outputs: three Main Outputs and one Aux Output.

Main Outputs

- XLR connector, balanced, AES/EBU professional format, 24-bits
- BNC connector, un-balanced, AES/EBU professional format, 24-bits, compatible with most SPDIF inputs
- Optical TOSLINK connector, multi-format (AES professional, ADAT, ADAT S/MUX II & IV), 24-bits

Aux Output

- BNC connector, AES Professional format, 16 or 24-bits

All of the outputs are controlled by the front-panel Mode Switch. The status of these outputs is shown in the Mode Display adjacent to the Mode Switch.

Three of the outputs are Main Outputs and always operate at 24-bits. The Main Outputs may be synchronized to an external clock reference or may be controlled by the internal clock. The Optical Output has two modes of operation; AES/EBU and ADAT. The ADAT mode supports ADAT (44.1 and 48 kHz), ADAT S/MUX^2 (88.2 and 96 kHz), and ADAT S/MUX^4 (176.4 and 192 kHz).

The Aux Output can operate asynchronously at 44.1 or 48 kHz with a TPDF-dithered 16-bit word length. The Aux Output is provided as a convenience for making safety backups or demo recordings to low-resolution 16-bit recorders (i.e. CDR or DAT). If this low-resolution function is not needed, the Aux Output can be set to mirror the high-resolution Main Outputs.

AES/EBU XLR Output

This output uses a gold-pin Neutrik™ male XLR connector. The output is balanced and has an output impedance of 110 Ohms. This output is DC-isolated, transformer-coupled, current-limited, and diode-protected. It is designed to drive standard 4 Vpp AES signals into a 110 Ohm load. Use 110 Ohm digital cable when connecting this output to other devices. The use of analog audio cables may cause data transmission errors.

- Data Format = AES/EBU professional format
- Word Length = 24 bits
- Sample Rate = 44.1, 48, 88.2, 96, 176.4, or 192 kHz
- Clock Source = Internal or external

Optical Output

The Optical Output has four modes of operation; AES/EBU, ADAT, ADAT S/MUX^2, and ADAT S/MUX^4. The ADAT LED on the front panel is illuminated whenever any of the ADAT Modes are active. S/MUX^2 and S/MUX^4 are automatically enabled if required to support the selected sample rate. S/MUX^2 is active at 88.2 or 96 kHz, S/MUX^4 is active at 176.4 or 192 kHz.

The Optical Output uses what is often called a TOSLINK, Type FO5, or 5 mm optical connector. The ADC1 uses a special high-bandwidth version that supports AES/EBU digital audio at sample rates up to 192 kHz. Please note that many optical inputs cannot support AES/EBU or SPDIF digital audio at sample rates above 48 kHz, others are limited to 96 kHz. A few products (such as the Benchmark DAC1) support 192 kHz optical inputs. Please note that high-bandwidth optical transmitters and receivers are not required for ADAT, ADAT S/MUX^2, or even ADAT S/MUX^4.

AES/EBU Optical Output Mode

- Data Format = AES/EBU professional format
- Word Length = 24 bits
- Sample Rate = 44.1, 48, 88.2, 96, 176.4, or 192 kHz
- Clock Source = Internal or external

ADAT Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 44.1 or 48 kHz
- Clock Source = Internal or external
- ADAT channel assignments: 1 = Left, 2 = Right, 3-8 = muted

ADAT S/MUX[2] Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 88.2 or 96 kHz
- Clock Source = Internal or external
- ADAT channel assignments *: 1 = Left a, 2 = Left b, 3 = Right a, 4 = Right b, 5-8 = muted

* a, and b are successive samples

ADAT S/MUX[4] Optical Output Mode:

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 176.4 or 192 kHz
- Clock Source = Internal or external
- ADAT channel assignments **: 1 = Left a, 2 = Left b, 3 = Left c, 4 = Left d, 5 = Right a, 6 = Right b, 7 = Right c, 8 = Right d

** a, b, c, and d are successive samples

SPDIF/AES BNC Main and Aux Outputs

The two BNC coaxial digital outputs use female BNC connectors. These connectors are securely mounted directly to the rear panel. These are 1 Vpp unbalanced outputs with 75-Ω source impedances. Outputs are DC-isolated, transformer-coupled, current-limited, and diode-protected. Use 75 Ohm coaxial cable when connecting these outputs to other devices. The use of 50 Ω coax is not recommended and may cause data transmission errors.

Many customers are more familiar with consumer-style RCA-equipped SPDIF digital interfaces. The ADC1 ships with BNC-to-RCA adapters. These adapters allow easy interfacing with consumer-style digital interfaces. BNC to RCA coaxial cords are also available from Benchmark.

BNC connectors are specified by the AES3-id and SMPTE 276M standards for 75-Ω 1 Vpp digital audio signals and are commonly used in video production facilities and other professional audio applications. RCA connectors are specified by IEC 609588-3 for 75-Ω 0.5 Vpp consumer-format digital audio signals (commonly known as SPDIF). We have chosen to comply with the professional standards because the BNC connectors lock and are generally more reliable than RCA connectors. Compliance with the 1 Vpp digital audio standards increases the reliability of digital connections, and often allows increased transmission distances.

Main BNC Output

This digital data at this output is identical to that of the Main XLR Digital Output.

- Data Format = AES/EBU professional format
- Word Length = 24 bits
- Sample Rate = 44.1, 48, 88.2, 96, 176.4, or 192 kHz
- Clock Source = Internal or external

Aux Output

This BNC digital output has two signals available to it. The first is a 16-bit TPDF auxiliary output for use with low-resolution devices. The second signal is the Main digital output and is identical to the data available at the other Main digital outputs.

- Data Format = AES/EBU professional format
- Word Length = 16 bits TPDF dithered, or 24 bits
- Sample Rate = 44.1 or 48 at 16-bits, 44.1, 48, 88.2, 96, 176.4, or 192 kHz at 24-bits
- Clock Source = Internal at 16-bits, internal or external at 24-bits

Word Clock Reference Output

This output provides a Word Clock signal for use with downstream components.

AC Power Entry Connector

The AC power input uses a standard IEC type connector. Within the USA and Canada, the ADC1 ships with a power cord. In other locations, a location-specific IEC style power cord may be purchased from a local source (including a local Benchmark dealer).

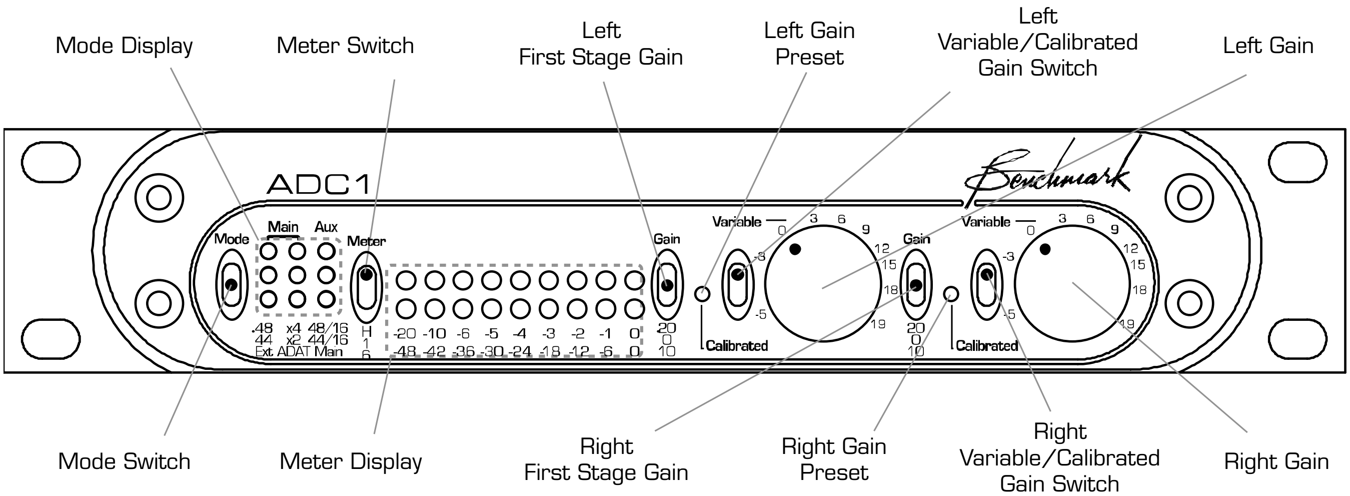
Fuse Holder

The fuse holder is built into a drawer next to the IEC power connector. The drawer requires two 5 x 20 mm 250 V Slo-Blo® Type fuses. The drawer includes a voltage selection switch with two settings: 110 and 220. Both settings use a 0.5 Amp fuse.

The AC input has a very wide input voltage range and can operate over a frequency range of 50 to 60 Hz. At 110, the ADC1 will operate normally over a range of 95 to 140 VAC. At 220, the ADC1 will operate normally over a range of 190 to 285 VAC.

CAUTION: ALWAYS REPLACE THE FUSES WITH THE CORRECT SIZE AND TYPE.

Operation



Mode Switch and Display

The ADC1 can be programmed to function in a variety of conversion modes, including sample rates, bit depths, and output formats, using internal and/or external clock sources. This programming is all done through the Mode Switch. The Mode Display shows the selected mode in a concise format.

The Mode Switch is a momentary toggle switch. There are two ways of operating the mode switch:

1. Press
2. Press and Hold

Pressing the Mode Switch momentarily and then releasing it results in a particular change to the ADC1 conversion mode, while pressing and holding the switch results in a different change.

To program the conversion mode

- Press the Mode Switch up repeatedly to cycle through the clock source and sample rate options for the Main Outputs.
- Press the Mode Switch down repeatedly to to cycle through the sample rate and bit depth options for the Aux Output.
- Press and hold the Mode Switch down for approximately 3 seconds to switch between AES/EBU and ADAT mode for the Optical Output.
- Press and hold the Mode Switch up for approximately 3 seconds to reset the ADC1 to Factory Default settings.

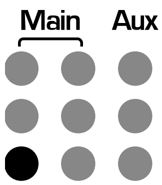
Details about all of these actions follow.

Programming the Outputs

Pressing up repeatedly on the mode switch cycles through the clock source and sample rate options for the Main Outputs. The Main Outputs can be set to operate at a fixed frequency using the internal clock source, or they can be set to follow and lock to an external clock source.

Locking to an External Clock Source

The ADC1 can sync to a variety of external clock sources, including Word Clock, Super Clock, AES, and SPDIF. Once the ADC1 acquires sync, it will perform conversion at the sample rate of the external clock.



Off = Internal Sync

On = Locked to External Sync

Flash = External Sync Selected but Not Locked

The bottom left LED in the Mode Display is the Ext Indicator. It shows that the ADC1 is locked to an external clock source. If the Ext LED is off, then the ADC1 is set to operate at a fixed sample rate using the internal clock source. If the Ext LED is on, the ADC1 is locked to an external clock. When locked, the Mode Display will indicate the sample rate. The ADC1 will automatically switch sample rates in response to changes in the reference sample rate. If the Ext LED is flashing, then the ADC1 is set to sync to an external clock source, but the ADC1 has not acquired a lock. The ADC1 should lock in less than 5 seconds. If the Ext LED flashes for more than 5 seconds, there is something wrong with the clock reference. Check the connections to the ADC1 Ref Input. The ADC1 will lock to AES, SPDIF, WC, or Super Clock and is very tolerant of low-level low-quality reference signals.

To synchronize with an external clock source

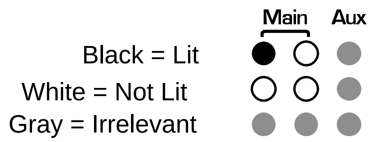
- Press up repeatedly on the Mode Switch, cycling through the Main Output modes until the lower left Ext LED is either on or flashing.

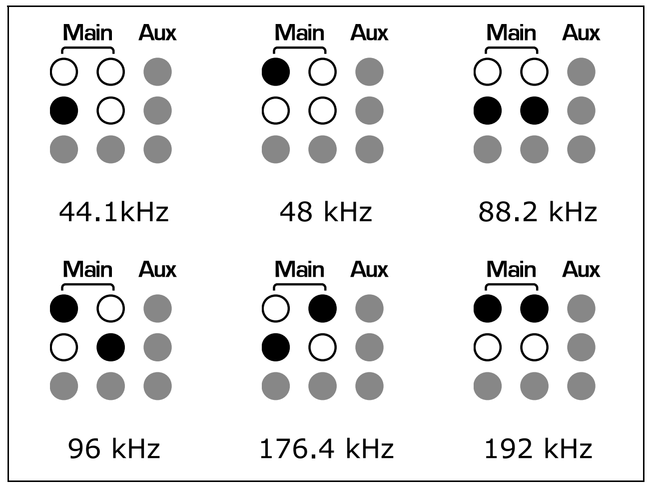
Selecting a Fixed Frequency Using the Internal Clock Source

The ADC1 can be programmed to convert at a fixed frequency using an internal clock source. The following sample rate frequencies are available: 44.1, 48, 88.2, 96, 176.4, and 192 kHz. The ADC1 External Clock Input is ignored when the internal clock source is selected.

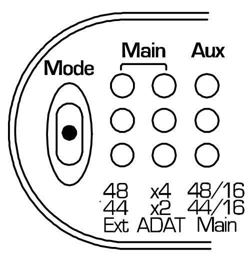
To select a fixed sample frequency on the Main Outputs

- Press up repeatedly on the Mode Switch to cycle through the available sample frequencies until the four LEDs in the upper left of the Mode Display match one of the diagrams below.





Reading Sample Rates off of the Mode Display



Column one of the display has a “44” LED and a “48” LED. These indicate sample rates of 44.1 kHz and 48 kHz respectively. Column two has an “X2” LED and an “X4” LED. These indicate 2x or 4x multipliers. Multiply the sample rate shown in column one by the multiplier shown in column two. For example, if the 44 and X2 LEDs are on, the sample rate is 88.2 kHz (44.1 x 2 = 88.2).

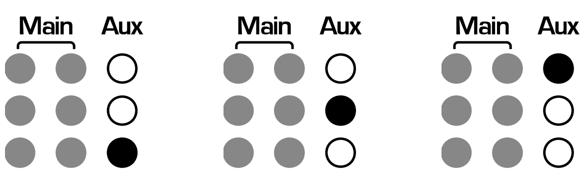
Programming the Aux Output

The Aux Output can be programmed to mirror the Main Outputs (bit for bit), or it can provide an independent low-resolution copy of the converted signal, at an independent sample rate. Column three of the Mode Indicator displays the Aux Output mode setting.

Note that no matter how the Aux Output is programmed it does not affect the Main Outputs in any way.

To program the Aux Output

Press down on the Mode Switch repeatedly to cycle through the Aux Output mode settings until the right-hand column of LEDs in the Mode Display matches the desired mode based on the diagrams below.



Exact Copy of Main Outputs (24-bit)	44.1 kHz 16-bit	48 kHz 16-bit

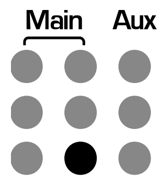
ADAT or AES/EBU on the Optical Output

The Optical Output (on of the three Main Outputs) can provide either AES/EBU or ADAT format. The bottom LED in the middle column of LEDs indicates what mode the Optical Output is in.

When ADAT is active, S/MUX is automatically enabled at all 2X and 4X sample rates (88.2 kHz, 96 kHz, 176.4 kHz, and 192 kHz).

To select between ADAT or AES/EBU on the Optical Output

Press and hold the Mode Switch down until the Optical Output mode LED matches the desired mode based on the diagram below.



Off = AES/EBU on Optical Output

On – ADAT on Optical Output

Resetting the ADC1 to Factory Default Settings

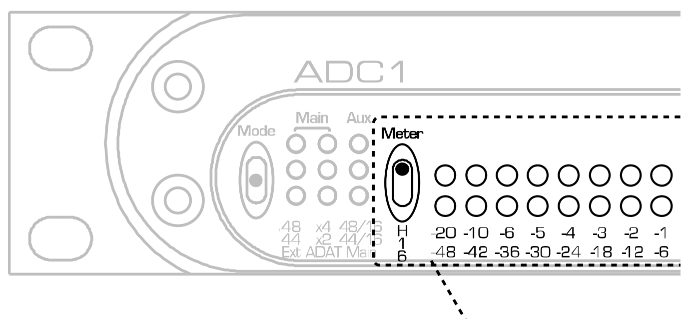
The ADC1 can be easily reset to Factory Default settings.

To reset the ADC1 to Factory Default settings

- Press and hold the Mode Switch up for approximately 3 seconds.

Meter Display

The ADC1 is equipped with a multi-function 9-segment LED meter. The Meter Switch selects either a 6 dB/step or 1 dB/step scale and controls the peak-hold function. Metering is fully-digital and is post conversion for absolute accuracy. The units are dBFS (dB below the level of a full-scale sine wave, or more simply, dB below digital clip).



Meter Switch and Meters

Time constants are built into the meters so that all transient peaks can be observed easily. If a transient peak having a duration as short as one digital sample occurs, an LED will be illuminated, and will stay illuminated long enough to be observed by the human eye.

A peak indication mimics the action of the needle on a peak-reading analog meter, while the remaining LEDs will follow the instantaneous level of the audio.

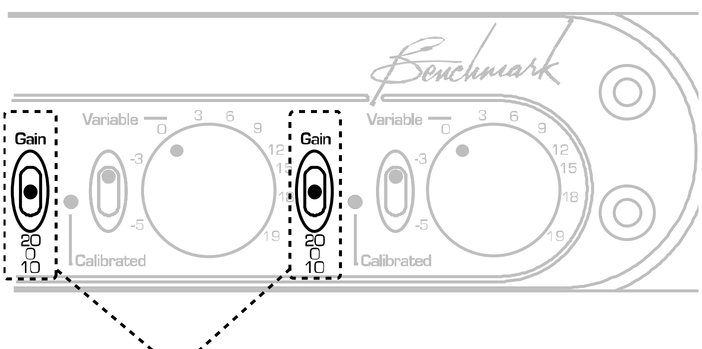
The red 0 LED indicates that a full-scale digital code has been reached and that digital clipping has occurred. Full-scale events as short as one digital sample, will light the 0 LED. Short single-sample digital clipping events are often audible, and all 0 dBFS events should be avoided.

The ADC1 has a very large dynamic range (especially when operating at 24-bit output word lengths). It is wise to use some of this dynamic range to provide more headroom as insurance against clipping. Leave some extra headroom between your highest anticipated peak and the red 0 dBFS LED.

To select the meter scale and peak hold function

- Set the Meter Switch to "H" (up) to enable the Peak Hold function and set the scale to 1 dB/step.
- Set the Meter Switch to "1" (center) to disable the Peak Hold function and set the scale to 1 dB/step.
- Set the Meter Switch to "6" (down) to disable the Peak Hold function and set the scale to 6 dB/step.

Adjusting Input Gain



Input Gain Stages

First Stage Gain

Each channel on the ADC1 is equipped with a 3-position first-stage gain switch. The first gain stage provides exceptional noise performance at gains of 0 dB, +10 dB, or +20 dB. This stage is followed by a second-stage having a continuously variable gain range of -1.3 dB to +22 dB. This gain structure provides ultra-high performance at any gain setting between -1.3 dB and +42 dB. The higher gain settings will allow direct connections from many instrument pickups (no DI box required).

To select the first stage gain

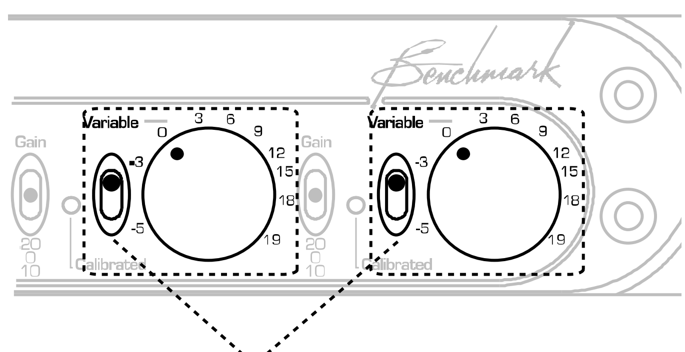
- Set the Gain Switch to "0" (center) to select 0 dB gain (unity gain) for the first-stage.
- Set the Gain Switch to "10" (down) to select 10 dB gain for the first-stage.
- Set the Gain Switch to "20" (up) to select 20 dB gain for the first-stage.

Second Stage Gain Controls

The second gain stage of each channel has a 41-detent Gain Control Knob, and a 10-turn Gain Calibration Trimmer. Each channel also has a 2-position Second-Stage Gain Switch. The switch selects either the Gain Control Knob or the Gain Calibration Trimmer. Both controls have a useable range of approximately -1.3 dB to +22 dB.

To use the Gain Control Knob to adjust second stage gain

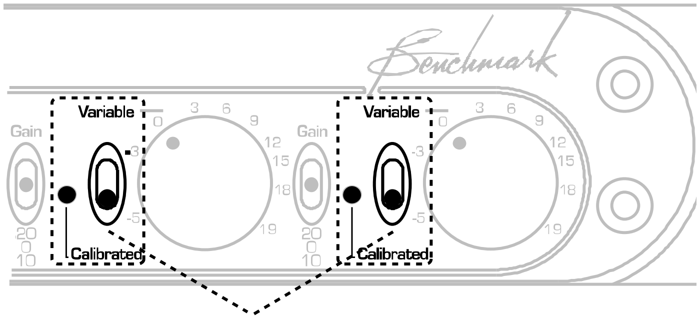
Set the Secondary Gain Switch to Variable (up) as shown in the diagram below.



Up Position

To use the Gain Calibration Trimmer to adjust second stage gain

Set the Secondary Gain Switch to Calibrated (down) as shown in the diagram below.



Rack Mounting

To enable rack mounting, the front panel of the ADC1 has rack-mount holes that are machined to conform to standard rack mount dimensions. The width of the ADC1 panel is exactly ½ that of a standard 19" panel. The ADC1 is one rack unit high. Either ear of the ADC1 can be mounted directly to a standard 19" rack. A machined junction block connects the other ear to a ½ width blank panel, another ADC1, a DAC1, or other ½ width Benchmark products. When joined, the two units form a single rigid 19" panel that can be installed in any standard 19" rack.

Using ADAT S/MUX

Proper S/MUX Identification is a Must

S/MUX² allows recording 4 channels at 88.2 or 96 kHz using a standard 8-channel 44.1 or 48 kHz ADAT recorder. S/MUX⁴ allows recording 2 channels at 176.4 or 192 kHz using a standard 8-channel 44.1 or 48 kHz ADAT recorder. In either case it is important to identify S/MUX recordings so that they can be properly decoded upon playback. Failure to properly decode an S/MUX recording will add unwanted artifacts to the audio. The severity of these artifacts is a function of the high-frequency content of the original digital audio signal, and may range from inaudible to very objectionable. This variation in severity can make it difficult to accurately spot a problem just by listening to a portion of the recording.

An ADAT S/MUX² recording will have pairs of nearly identical tracks (1≈2, 3≈4, 5≈6, and 7≈8). Unfortunately this can be mistaken for 4 stereo pairs at half of the original sample rate. There is no substitute for proper labeling. This labeling should include the sample rate of the recording.

An ADAT S/MUX⁴ recording is somewhat easier to identify because it will have groups of 4 channels that are nearly identical (1≈2≈3≈4, and 5≈6≈7≈8). In error, S/MUX⁴ could be played at ¼ of its original sample rate, and sound almost normal. S/MUX⁴ could also be mistaken for S/MUX² and could be played at ½ of its original sample rate. Please note that these changes in sample rate will not alter the pitch of the audio but will introduce errors. These errors may not be discovered until it is too late.

S/MUX Must be Decoded Before Digital Processing

No DSP process should be applied to an S/MUX signal before it is decoded. S/MUX must be decoded before it reaches the internal processing in a DAW or a digital console. Many such devices include S/MUX decoders at their digital interfaces. These decoders must be properly enabled for S/MUX and must be disabled for standard ADAT inputs.

Sample Rate is the Key that Controls S/MUX

Most devices (including the ADC1) automatically enable and disable S/MUX in response to changes in sample rate. Therefore it is essential that all S/MUX equipped A/D converters, D/A converters, digital consoles, digital audio workstations, and digital processing devices be set to identical sample rates. There is one exception to this rule: A non-S/MUX ADAT recorder can be connected to an S/MUX interface, but the recorder must be set at ½ (S/MUX²) or ¼ (S/MUX⁴) of the actual sample rate.

S/MUX should not be used for Sample Rate Conversion

If two devices are connected with an ADAT S/MUX interface and the devices are set to different sample rates, a crude form of sample rate conversion will occur. For example, if an A/D converter is set to 96 kHz, and it feeds a digital console that is set to 48 kHz, the system will appear to down convert from 96 kHz to 48 kHz. This would be a useful feature if the digital filtering was correct. The problem is that this ad-hoc sample rate converter is lacking the low-pass filter that prevents aliasing.

UltraLock™ … What is It?

Accurate 24-bit audio conversion requires a very low-jitter conversion clock. Jitter can very easily turn a 24-bit converter into a 16-bit converter (or worse). There is no point in buying a 24-bit converter if clock jitter has not been adequately addressed.

Jitter is present on every digital audio interface. This type of jitter is known as *interface jitter* and it is present even in the most carefully designed audio systems. Interface jitter accumulates as digital signals travel down a cable and from one digital device to the next. If we measure interface jitter in a typical system we will find that it is 10 to 10,000 times higher than the level required for accurate 24-bit conversion. Fortunately, this interface jitter has absolutely no effect on the audio *unless* it influences the conversion clock in an analog-to-digital converter (ADC) or in a analog-to-digital converter (DAC).

Many converters use a single-stage Phase Lock Loop (PLL) circuit to derive their conversion clocks from AES/EBU, Word Clock, or Super Clock reference signals. Single-stage PLL circuits provide some jitter attenuation above 5 kHz but none below 5 kHz. Unfortunately, digital audio signals often have their strongest jitter components at 2 kHz. Consequently, these converters can achieve their rated performance only when driven from very low jitter sources and through very short cables. It is highly unlikely that any converter with a single-stage PLL can achieve better than 16 bits of performance in a typical installation. Specified performance may be severely degraded in most installations.

Better converters usually use a two-stage PLL circuit to filter out more of the interface jitter. In theory, a two-stage PLL can remove enough of the jitter to achieve accurate 24-bit conversion (and some do). However, not all two-stage PLL circuits are created equal. Many two-stage PLLs do not remove enough of the low-frequency jitter. In addition, two-stage PLL circuits often require several seconds to lock to an incoming signal. Finally, a two-stage PLL may fail to lock when jitter is too high, or when the reference sample frequency has drifted.

UltraLock™ converters exceed the jitter performance of two-stage PLL converters, and are free from the slow-lock and no-lock problems that can plague two-stage PLL designs. UltraLock converters are 100% immune to interface jitter under all operating conditions. No jitter-induced artifacts can be detected using an Audio Precision System 2 Cascade test set. Measurement limits include detection of artifacts as low as –140 dBFS, application of jitter amplitudes as high as 12.75 UI, and application of jitter over a frequency range of 2 Hz to 200 kHz. Any AES/EBU signal that can be decoded by the AES/EBU receiver will be reproduced without the addition of any measurable jitter artifacts.

The ADC1, DAC-104 and the ADC-104 employ Benchmark's new UltraLock technology to eliminate all jitter-induced performance problems. UltraLock isolates the conversion clock from the digital audio interface clock. Jitter on a DAC digital audio input, or an ADC reference input can never have *any* measurable effect on the conversion clock of an UltraLock converter. In an UltraLock converter, the conversion clock is never phase-locked to a reference clock. Instead the converter oversampling-ratio is varied with extremely high precision to achieve the proper phase relationship to the reference clock. Interface jitter cannot degrade the quality of the audio conversion. Specified performance is consistent and repeatable in any installation!

How does conversion clock jitter degrade converter performance?

Problem #1

Jitter phase modulates the audio signal. This modulation creates sidebands (unwanted tones) above and below every tone in the audio signal. Worse yet, these sidebands are often widely separated from the tones in the original signal.

Jitter-induced sidebands are not musical in nature because they are not harmonically related to the original audio. Furthermore, these sidebands are poorly masked (easy to hear) because they can be widely separated above and below the frequencies of the original audio tones. In many ways, jitter induced distortion resembles intermodulation distortion (IMD). Like IMD, jitter induced distortion is much more audible than harmonic distortion, and more audible than THD measurements would suggest.

Jitter creates *new audio* that is not harmonically related to the original audio signal. This new audio is unexpected and unwanted. It can cause a loss of imaging, and can add a low and mid frequency "muddiness" that was not in the original audio.

Jitter induced sidebands can be measured using an FFT analyzer.

Problem #2

Jitter can severely degrade the anti-alias filters in an oversampling converter. This is a little known but easily measurable effect. Most audio converters operate at high oversampling ratios. This allows the use of high-performance digital anti-alias filters in place of the relatively poor performing analog anti-alias filters. In theory, digital anti-alias filters can have extremely sharp cutoff characteristics, and very few negative effects on the in-band audio signal. Digital anti-alias filters are usually designed to achieve at least 100 dB of stop-band attenuation. But, digital filters are designed using the mathematical assumption that the time interval between samples is a constant. Unfortunately, sample clock jitter in an ADC or DAC varies the effective time interval between samples. This variation alters the performance of these carefully designed filters. Small amounts of jitter can severely degrade stop-band performance, and can render these filters useless for preventing aliasing.

The obvious function of a digital anti-alias filter is the removal of audio tones that are too high in frequency to be represented at the selected sample rate. The not-so-obvious function is the removal of high-frequency signals that originate inside the converter box, or even originate inside the converter IC. These high-frequency signals are a result of crosstalk between digital and analog signals, and may have high amplitudes in a poorly designed system. Under ideal (low jitter) conditions, a digital anti-alias filter may remove most of this unwanted noise before it can alias down into lower (audio) frequencies. These crosstalk problems may not become obvious until jitter is present.

Stop-band attenuation can be measured very easily by sweeping a test tone between 24 kHz and at least 200 kHz while monitoring the output of the converter.

Put UltraLock converters to the test

We encourage our customers to perform the above tests on UltraLock converters (or let your ears be the judge). There will be absolutely no change in performance as jitter is added to any digital input on an UltraLock converter.

Try the same tests on any converter using conventional single or two-stage PLL circuits. Tests should be performed with varying levels of jitter and with varying jitter frequencies. The results will be very enlightening. Jitter related problems have audible (and measurable) effects on ADC and DAC devices. Practitioners of Digital Audio need to understand these effects.

Is it possible to eliminate all of the effects of jitter in an entire digital audio system?

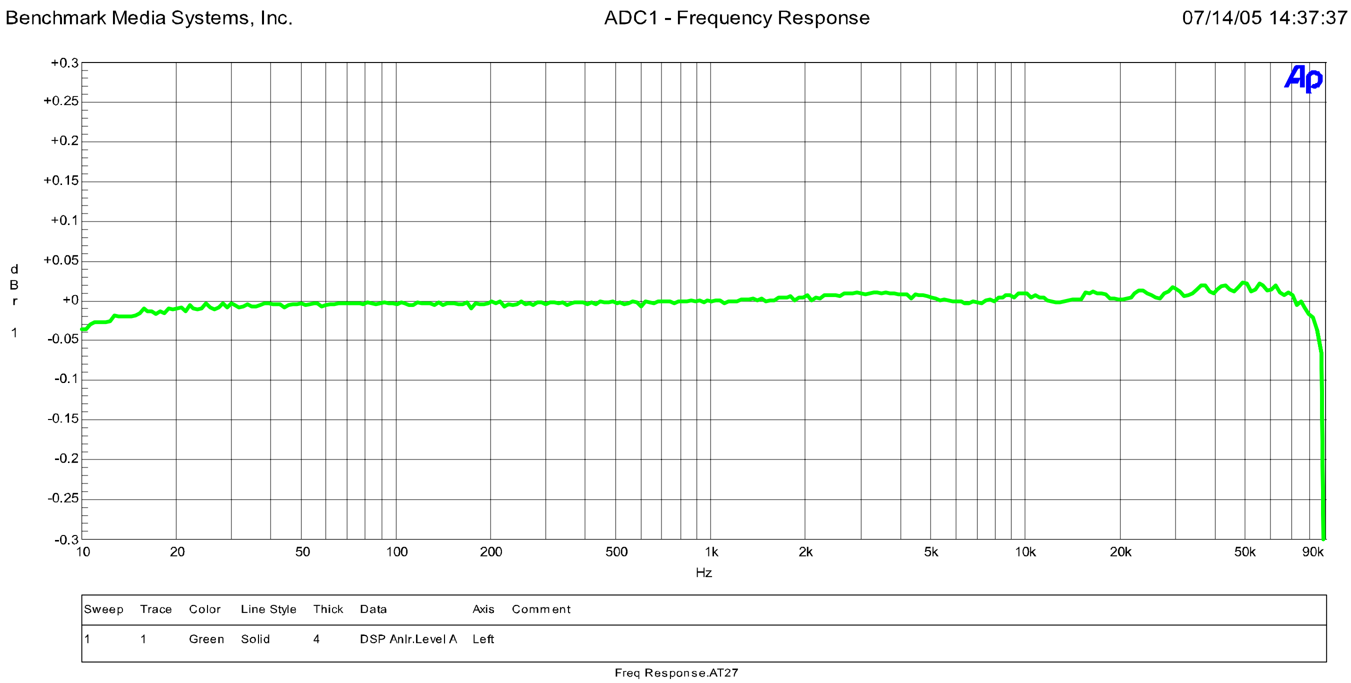
Interface jitter will accumulate throughout even the most carefully designed digital audio system. Fortunately, interface jitter can only degrade digital audio if it affects the sampling circuit in an analog-to-digital or analog-to-digital converter. Any attempt to cure jitter outside of an ADC or DAC will prove expensive and, at best, will only partially reduce jitter-induced artifacts. Dedicated clock signals (word clock, and super clock, etc.) are often distributed to A/D converters and D/A converters in an attempt to reduce jitter. Again, these are only partial solutions because jitter even accumulates in these clock distribution systems. Furthermore, a poor quality master clock generator can degrade the performance of the entire system (if converter performance is dependent upon reference clock quality. Jitter free ADCs and DACs are the only true insurance against the ill effects of jitter. UltraLock converters are jitter immune under all operating conditions (they will never add audible jitter induced artifacts to an audio signal).

What UltraLock converters cannot do

UltraLock converters cannot undo damage that has already been done. If an ADC with a jitter problem was used to create a digital audio signal, then there is nothing that can be done to remove the damage. Jitter-induced sidebands are extremely complex and cannot be removed with any existing audio device. It is therefore important to attack jitter at both ends of the audio chain. The ADC1 is a great start, as it will allow accurate assessment of various A/D converters. It is impossible to evaluate ADC performance without a good DAC. The consistent performance delivered by the ADC1 eliminates one major variable: jitter.

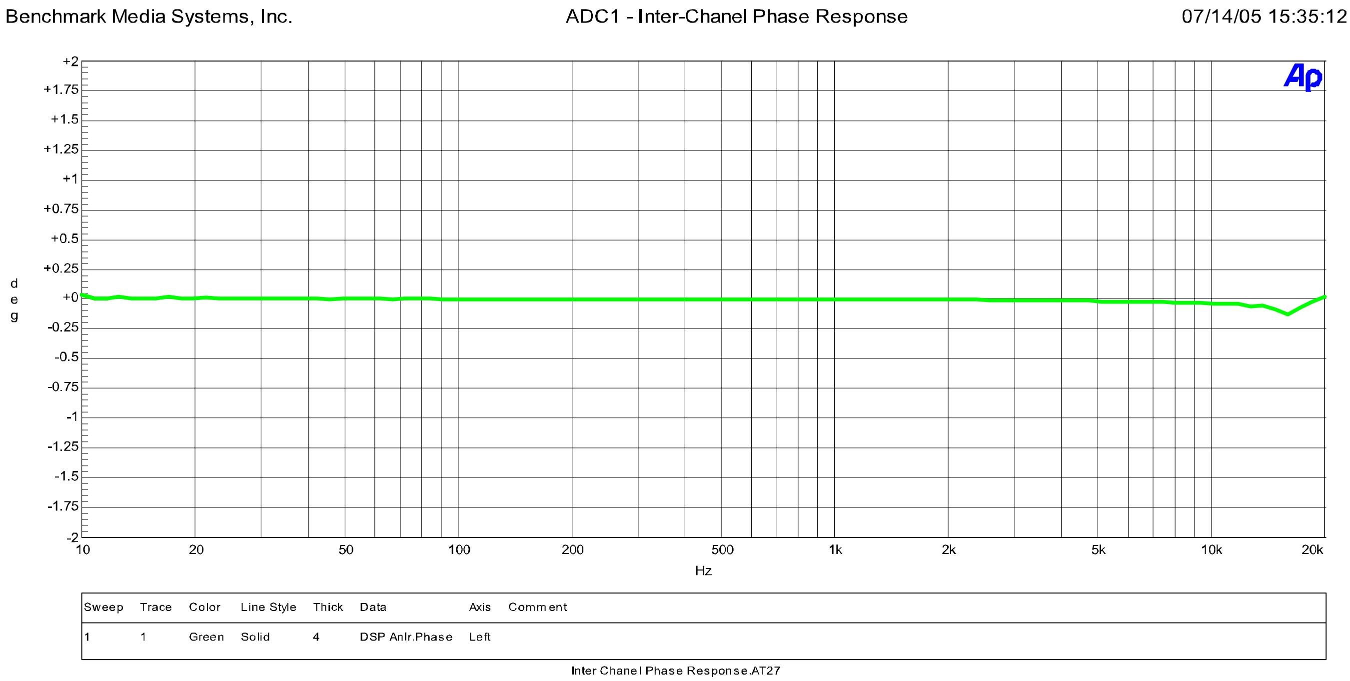
Performance

Frequency Response



The above graphs show the frequency response of the ADC1 when it is operating at a 192-kHz sample rate. Note that the amplitude response is down by less than 0.05 dB at 10 Hz and 80 kHz. The bass response extends well below the 10-Hz limitation of the measurement equipment, and the high-frequency analog response extends well above the 96 kHz bandwidth of 192 kHz digital audio.

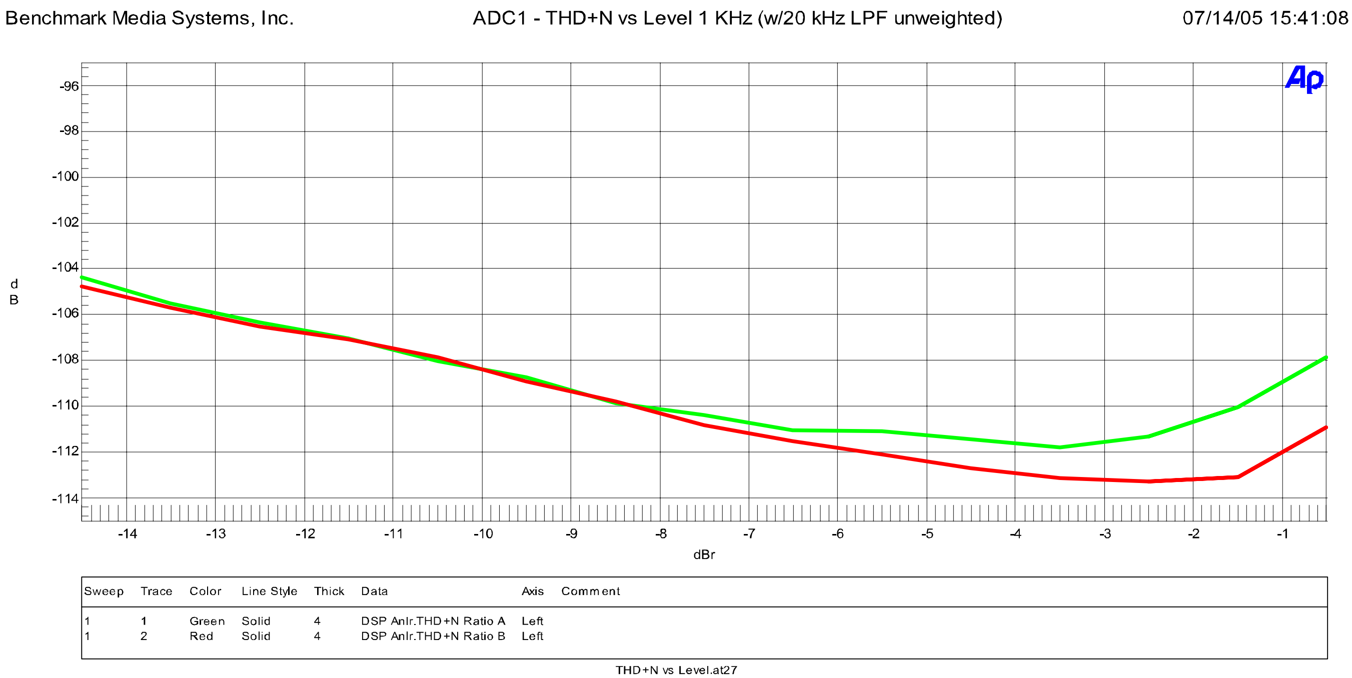
Inter-Channel Phase Response



This graph shows that the differential phase is significantly better than ± 0.25º from 10 Hz to 20 kHz.

THD+N vs. Level, 1 KHz

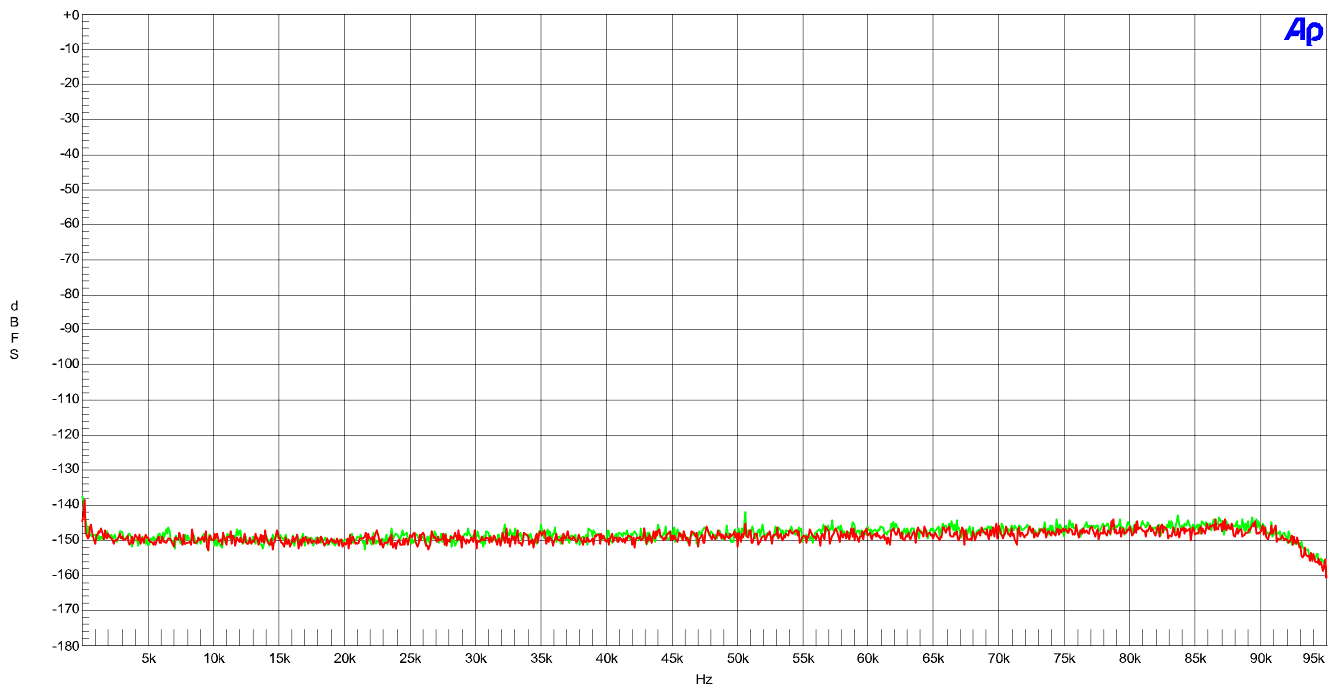
w/20 kHz LPF unweighted



Below −4 dBFS, distortion is lower than the noise floor of the converter. Above −3 dBFS, distortion reaches a maximum value of only −107 dBFS.

32K B-H FFT, Idle Channel Noise

Benchmark Media Systems, Inc. 32K B-H FFT Analysis of Idle Channel Noise 07/14/05 14:11:42

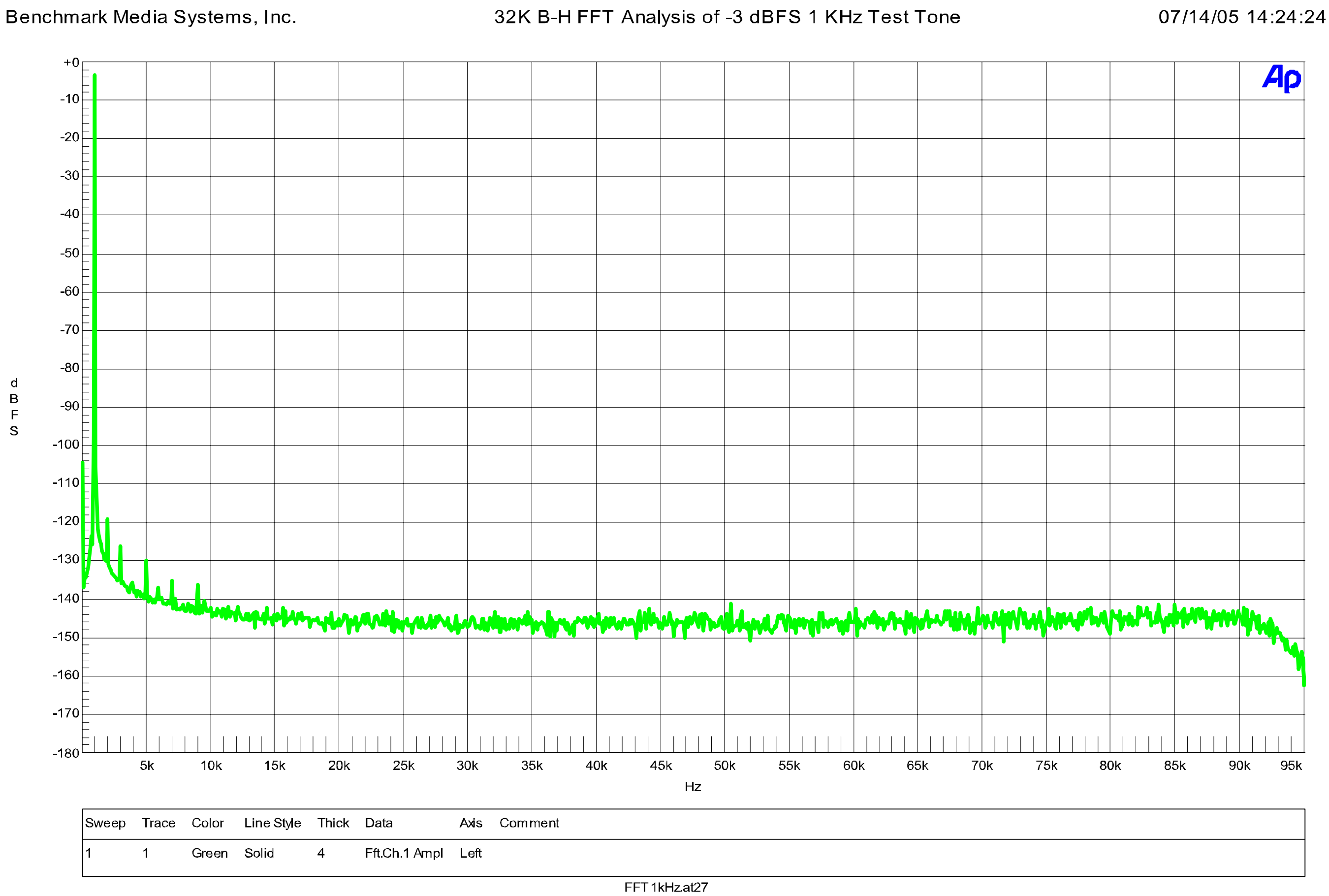


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left	
1	2	Red	Solid	1	Fft.Ch.2 Ampl	Left	

B - FFT Idle Chanel Noise.at27

The above graph demonstrates that the ADC1 is free from idle tones and clock crosstalk. The highest spurious tone measures –128 dBFS and is AC line related hum. The highest non-line related tone measures –135 dBFS.

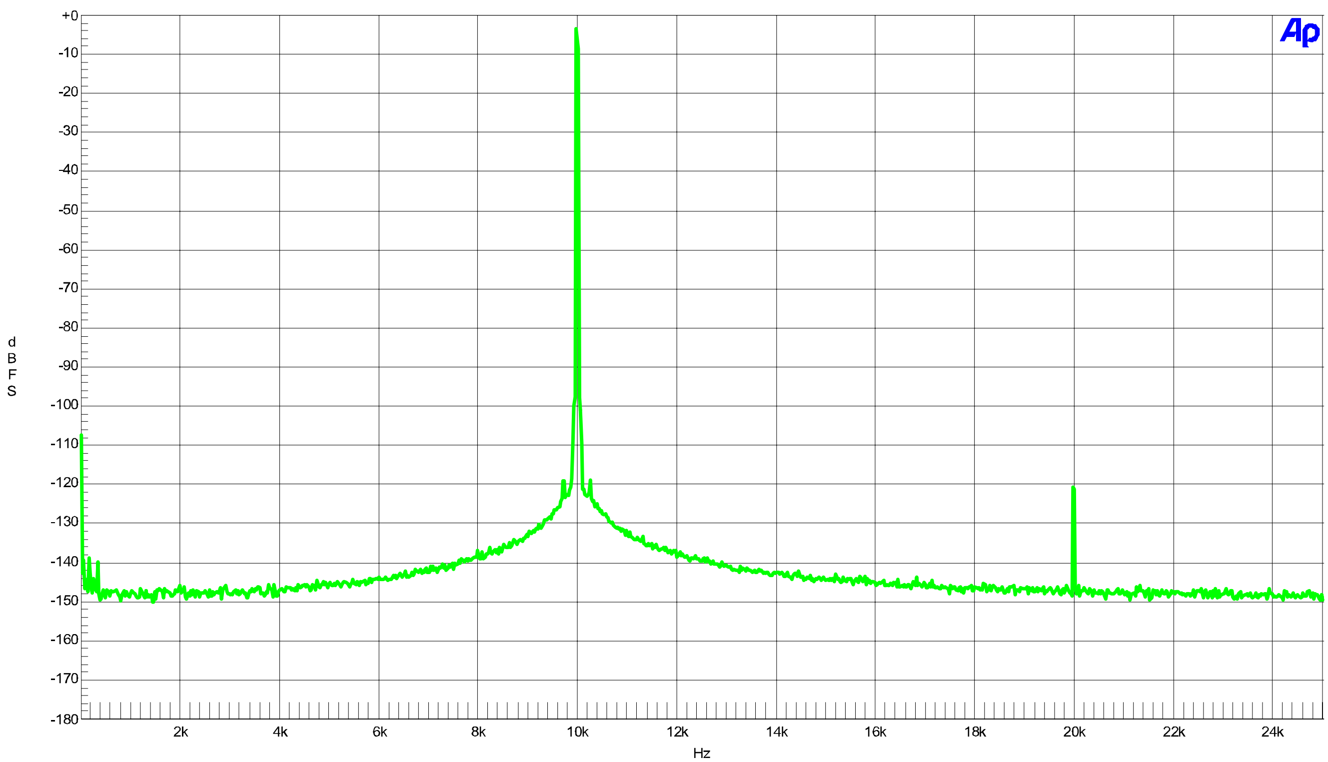
32K B-H FFT, -3 dBFS, 1 KHz



The above FFT plot shows that the ADC1 has very little harmonic distortion. Distortion is exceptionally low and is dominated by 2nd harmonic distortion. Note the near absence of spurious tones.

32K B-H FFT, -3 dBFS, 10 KHz

Benchmark Media Systems, Inc. 32K B-H FFT Analysis of -3 dBFS 10 KHz Test Tone 07/14/05 15:38:55



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left	

FFT 10kHz.at27

The above FFT plot shows that the ADC1 is free from jitter-induced sidebands. Any jitter present at the conversion sampling circuit would produce sidebands equally spaced above and below the 10 kHz test tone. The tone at 20 kHz is due to second harmonic distortion, and measures almost 120 dB below full scale. Note the near absence of spurious tones.

Specifications

Analog Audio Inputs

Number of Inputs (balanced)	2
Connector	Gold-Pin Neutrik™ female XLR
Impedance	200 kΩ
Sensitivity	-14dBu to +29 dBu (at 0 dBFS)

Clock Reference Input

Format	Auto-detect AES/EBU, Word Clock, and Super Clock (256x)
Impedance	75 Ω
Sensitivity	150 mV AES 200 mV Word Clock 750 mV Super Clock
Transformer Coupled	Yes
DC Blocking Capacitors	Yes
Transient and Over-Voltage Protection	Yes
Jitter Attenuation Method	Benchmark *UltraLock*™

Worldclock Reference Output

Impedance	75 Ω
Level	5 Vpp 2.5 Vpp into 75 Ω
Transformer Coupled	No
DC Blocking Capacitors	No
Transient and Over-Voltage Protection	Yes

Digital Audio Outputs

Number of Digital Outputs	1 XLR Main 1 TOSLINK Main 1 BNC Main 1 BNC Aux
Connectors	Gold-Pin Neutrik™ male XLR
Number of Audio Channels	2
Main Output Word Length	24 bits
Main Output Sample Frequencies	44.1, 48, 88.2, 176.4, or 192 kHz
Aux Output Word Length	16 or 24 bits
Aux Output Sample Frequencies	44.1, 48, 88.2, 176.4, or 192 kHz at 24 bits 44.1 or 48 at 16 bits
Impedance	110 Ω XLR 75 Ω BNC
Level	4 Vpp into 100 Ω XLR 1 Vpp into 75 Ω BNC
Transformer Coupled	Yes
DC Blocking Capacitors	Yes
Transient and Over-Voltage Protection	Yes

Audio Performance

Fs = 44.1 to 192 kHz, 20 to 20 kHz BW, 1 kHz test tone, 0 dBFS = +24 dBu (unless noted)

SNR – A-Weighted, 0 dBFS = +8 to +29 dBu	121 dB
SNR – Unweighted, 0 dBFS = +8 to +29 dBu	119 dB
SNR – A-Weighted at max gain, 0 dBFS = -14 dBu	108 dB
THD+N, 1 kHz at –1 dBFS	-102 dBFS, -101 dB, 0.00089%
THD+N, 1 kHz at –3 dBFS	-107 dBFS, -104 dB, 0.00063%
THD+N, 20 to 20 kHz test tone at –3 dBFS	-106 dBFS, -103 dB, 0.00071%
Frequency Response at Fs=192 kHz	-3 dB, +0 dB, 2 Hz to 92 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz -0.00 dB at 20 kHz -0.18 dB at 88 kHz -3 dB at 92 kHz -100 dB at 108 kHz
Frequency Response at Fs=96 kHz	-3 dB, +0 dB,1 Hz to 46 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz -0.00 dB at 20 kHz -0.10 dB at 44 kHz -3 dB at 46 kHz -108 dB at 54 kHz
Frequency Response at Fs=48 kHz	3 dB, +0 dB, 1 Hz to 23 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz -0.00 dB at 20 kHz -0.10 dB at 22 kHz -3 dB at 23 kHz, -110 dB at 27 kHz
Passband Ripple	+/- 0.008 dB
Crosstalk	-105 dB at 20 kHz -130 dB at 1 kHz -200 dB at 20 Hz

Jitter Tolerance (With no Measurable Change in Performance)	>12.75 UI sine, 100 Hz to 10 kHz > 3.5 UI sine at 20 kHz > 1.2 UI sine at 40 kHz > 0.4 UI sine at 80 kHz > 0.29 UI sine at 90 kHz > 0.25 UI sine above 160 kHz
Maximum Amplitude of Jitter Induced Sidebands	< -134 dB (measurement limit) (10 kHz 0 dBFS test tone, 12.75 UI sinusoidal jitter at 1 kHz)
Maximum Amplitude of Spurious Tones with 0 dBFS test signal	-130 dBFS
Maximum Amplitude of Idle Tones	-145 dBFS
Maximum Amplitude of AC line related Hum & Noise	-130 dBFS
Interchannel Differential Phase (Stereo Pair)	+/- 0.5 degrees at 20 kHz
Interchannel Differential Phase (Between ADC1 Units)	+/- 0.5 degrees at 20 kHz
Maximum Lock Time after Fs change	< 1 s for frequency lock < 5 s for phase lock
Mute on Sample Rate Change	Yes
Mute on Loss of External Clock	No
Mute on Lock Error	No
Mute on Receive Error	No
Soft Mute Ramp Up/Down Time	10 ms

Group Delay (Latency)

Delay (Analog Input to Digital Output)	1.20 ms at 44.1 kHz 1.09 ms at 48 kHz 0.75 ms at 88.2 kHz 0.67 ms at 96 kHz 0.63 ms at 176.4 kHz 0.59 ms at 192 kHz

LED Status Indicators

LED Location	Front Panel
Mode Indicators	9 green
Meter	14 green, 2 yellow, 2 red

AC Power Requirements

Input Operating Voltage Range (VAC RMS)	110 V setting – 95 V min, 140 V max 220 V setting – 190 V min, 285 V max
Frequency	50-60 Hz
Power	16 Watts Idle 16 Watts Typical Program 20 Watts Maximum
Fuses	5 x 20 mm (2 required) 110 V setting – 0.5 A 250 V Slo-Blo® Type 220 V setting – 0.5 A 250 V Slo-Blo® Type

Dimensions

Form Factor	½ Rack Wide, 1 RU High
Depth behind front panel	8.5” (216 mm)
Overall depth including connectors but without power cord or BNC-to-RCA adapter	9.33” (237 mm)
Width	9.5” (249 mm)
Height	1.725” (44.5 mm)

Weight

ADC1 only	3.6 lb.
ADC1 with power cord, 3 BNC-to-RCA adapters, extra fuses, and manual	4.9 lb.
Rack mount kit (blank panel, junction block, and rack-mount screws)	0.32 lb.
Shipping weight	7 lb.

Certificate Of Conformity

Diversified T.E.S.T. Technologies, Inc. has tested the product to the current appropriate standards and finds that the product is in compliance with those requirements.

EMC Directive:	**89/336/EEC**	
Generic Emissions Standard:	EN 61000-6-3: 2001	
Product Specific Emissions:	EN 55022 Class B	
Generic Immunity Standard:	EN 61000-6-1: 2001	
Immunity:	EN 61000-4-2	Electrostatic Discharge
	EN 61000-4-3	Radiated Susceptibility
	EN 61000-4-4	Electrical Fast Transient/Burst
	EN 61000-4-5	Surge
	EN 61000-4-6	Conducted Susceptibility
	EN 61000-3-2	Harmonic Current
	EN 61000-3-3	Voltage Fluctuations & Flicker
Low Voltage Directive:	**98/68/EEC**	
Standard:	EN 60950	(ITE) Information Technology Equipment

Manufacturer's Name:	Benchmark Media
Manufacturer's Address:	5925 Court Street Syracuse, NY 13026
Product:	Audio A-D Converter
Model Number:	ADC1

This Certificate of Compliance issued July 13, 2005 is valid for the test sample of the product specified above and that it conforms to the Directive(s) and Standard(s).

Signature:

Thomas P. Sims
President
Diversified T.E.S.T. Technologies, Inc.
PO Box 8, 556 Route 222
Groton, NY 13073
Phone: 607-898-4218
Fax: 607-898-4830

DIVERSIFIED T.E.S.T. TECHNOLOGIES, INC. CORPORATE SEAL 1991 NEW YORK

Warranty Information

Benchmark 1 Year Warranty

The Benchmark 1 Year Warranty

Benchmark Media Systems, Inc. warrants its products to be free from defects in material and workmanship under normal use and service for a period **of one (1) year from the date of delivery.**

This warranty extends only to the original purchaser. This warranty does not apply to fuses, lamps, batteries, or any products or parts that have been subjected to misuse, neglect, accident, modification, or abnormal operating conditions.

In the event of failure of a product under this warranty, Benchmark Media Systems, Inc. will repair, at no charge, the product returned to its factory. Benchmark Media Systems, Inc. may, at its option, replace the product in lieu of repair. If the failure has been caused by misuse, neglect, accident, or, abnormal operating conditions, repairs will be billed at the normal shop rate. In such cases, an estimate will be submitting before work is started, if requested by the customer.

Attempts to deliberately deface, mutilate, or remove the product's label will render this warranty void. Any ADC1 with a serial number greater than 00261 returned from the European Union for warranty repair must have the required RoHS logo on the product label; otherwise, repairs will be billed at the normal shop rate. Benchmark will not honor warranties for any products disingenuously purchased on the US or Canadian markets for sale outside the US or Canada.

The foregoing warranty is in lieu of all other warranties, expressed or implied, including but not limited to any implied warranty of merchantability, fitness or adequacy for any particular purpose or use. Benchmark Media Systems, Inc. shall not be liable for any special, incidental, or consequential damages, and reserves the right to charge this information without notice. This limited warranty gives the consumer-owner specific legal rights, and there may also be other rights that vary form state to state.

Benchmark Extended Warranty

The Benchmark Extended 5* Year Warranty

Benchmark Media Systems, Inc. optionally extends the standard one (1) year warranty to a period of **five (5)* years from the date of delivery.**

*For the extended warranty to become effective, the original purchaser must register the product at the time of purchase either by way of the prepaid registration card or through the product registration section of the Benchmark Media Systems, Inc. website. This optional warranty applies only to products purchased within the US and Canada and is extended only to the original purchaser.

Attempts to deliberately deface, mutilate, or remove the product's label will render this warranty void. Benchmark will not honor warranties for any products disingenuously purchased on the US or Canadian markets for export. The terms of the extended warranty are subject to change without notice. For products purchased outside the US and Canada, please refer to the Extended Two (2)** Year International Warranty.

The Benchmark's Extended 2** Year International Warranty

Benchmark Media Systems, Inc. optionally extends the standard one (1) year warranty to a period of **two (2)** years from the date of delivery.**

**For the extended warranty to become effective, the original purchaser must register the product at the time of purchase either by way of the prepaid registration card or through the product registration section of the Benchmark Media Systems, Inc. website. This optional warranty applies only to products purchased outside the US and Canada and is extended only to the original purchaser.

Attempts to deliberately deface, mutilate, or remove the product's label will render this warranty void. Benchmark will not honor warranties for any products disingenuously purchased on the US or Canadian markets for export. The terms of the extended warranty are subject to

Notes on Warranty Repairs

An RMA (return merchandise authorization) number, issued by our Customer Service Department, is required when sending products for repair.

They must be shipped to Benchmark Media Systems prepaid and preferably in their original shipping carton with the RMA number clearly visible on the exterior of the packaging. A letter should be included giving full details of the difficulty.

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...the measure of excellence!™